IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

In re Application of:	Olaf Zaencker
Serial No.:	10/076,108
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Group Art Unit:	2419
Confirmation No.:	8552
Examiner:	Duong, Duc T.
Title:	**METHOD AND ARRANGEMENT FOR TESTING THE TRANSMISSION SYSTEM AND METHOD FOR QUALITY OF A SPEECH TRANSMISSION**

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REPLACEMENT APPEAL BRIEF

Further to the Appeal Brief submitted on June 8, 2009, Appellant hereby submits this Replacement Appeal Brief in response to the Notification of Non-Complaint Appeal Brief mailed July 8, 2009.

APPELLANT'S BRIEF (37 C.F.R. § 41.37)

This brief is submitted in support of Appellant's notice of appeal from the decision dated December 8, 2008 of the Examiner finally rejecting claims 1-2 and 5-23 of the subject application.

I. REAL PARTY IN INTEREST

This application is currently owned by Siemens Aktiengesellschaft as indicated by an assignment recorded on May 6, 2002, in the Assignment Records of the United States Patent and Trademark Office at Reel 012879, Frame 0521.

II. RELATED APPEALS AND INTERFERENCES

There are no known appeals or interferences which will directly affect or be directly affected by or have a bearing on the Board's decision regarding this appeal.

III. STATUS OF CLAIMS

Claims 1-2 and 5-23 are pending in this application. Claims 3-4 were previously cancelled without prejudice or disclaimer. Claims 1-2, 5, and 8-21 stand rejected under a Final Office Action mailed December 8, 2008. Claims 6, 7, and 22-23 are allowable but objected to as they depend from claims that currently stand rejected. Appellant presents Claims 1-2, 5, and 8-21 for appeal. Appendix A shows all pending claims.

IV. STATUS OF AMENDMENTS

Subsequent to the Final Office Action mailed December 8, 2008, Appellant amended independent Claims 1 and 14-16 to overcome the Examiner's rejection based on 35 U.S.C. § 112. The Examiner rejected these claims noting that the recitation of "the enumeration" in Claims 1 and 14-16, and "the conneetion" in Claim 14 lacked sufficient antecedent basis. Thus, in Claims 1 and 14-16 Appellants changed "the enumeration" to "an enumeration." In the Advisory Action mailed February 2, 2009, the Examiner indicated that the proposed amendments will be entered, and that the rejection of Claims 1 and 14-16 based on 35 U.S.C. § 112 is overcome.

Although the Examiner indicated in the Advisory Aetion that the rejection under § 112 has been overcome, Appellant subsequently discovered that Claim 14 was accidentally not amended in response the Examiner's rejection based on the lack of antecedent basis for the recited limitation "the connection." Thus, Appellant amended Claim 14 in an amendment mailed June 5, 2009. This amendment changed one instance of "the connection" (in the last element of the claim) to "a connection." As of the filing of this Appeal Brief, this amendment has not yet been entered. Even though this amendment has not yet been entered, the listing of Claim 14 in Appendix A includes this change.

V. SUMMARY OF CLAIMED SUBJECT MATTER

Numerals in parentheses have been added and refer to the embodiment shown in FIG. 3 of the present application.

Independent Claim 1 relates to a method for testing the transmission quality of a bidirectional real speech transmission or multicast connection over an IP network (11) between a first VoIP endpoint (13a) and a second VoIP endpoint (13b). *See, e.g.*, ¶¶ 0045-0047, figs. 1, 3. The method includes, for example, transmitting Real-Time Transport Protocol (RTP) speech packets from the first VoIP endpoint (13a) to the second VoIP endpoint (13b), and vice versa. *See, e.g.*, ¶¶ 0045-0047, figs. 1, 3. Additionally, the method includes detecting an enumeration (19a) of the transmitted RTP speech packets from the first (13a) to the second (13b) VoIP endpoint as a first number, and an enumeration (19b) of the transmitted RTP speech packets from the second (13b) to the first (13a) VoIP endpoint as a second number. *See, e.g.*, ¶¶ 0047, 0050-0052, 0059, figs. 1, 3. This detection occurs over a predetermined time period (21) at a detection point on a transmission channel (15) between the first (13a) and second (13b) VoIP endpoints. *See, e.g.*, ¶¶ 0049, 0060, 0064-0065, fig. 3. The method further includes, for example, arithmetically processing (23) the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality. *See, e.g.*, ¶¶ 0051, 0053, 0065-0068, fig. 3.

Independent Claim 14 relates to a method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint. *See, e.g.*, ¶¶ 0045-

0047, 0054-0057, 0068, figs. 1, 3. The method recites limitations similar to those described above for Claim 1, and additionally includes routing the connection between the first (13a) and second (13b) VoIP endpoints based on the value representing the transmission quality that is output from the arithmetical processing. *See, e.g.*, ¶¶ 0054, 0056, 0068, fig. 1.

Likewise, independent Claim 15 relates to a method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint. *See, e.g.*, ¶¶ 0045-0047, 0054-0057, 0068, figs. 1, 3. The method recites limitations similar to those described above for Claim 1, and additionally includes setting transmission parameters based on the value representing the transmission quality that is output from the arithmetical processing. *See, e.g.*, ¶¶ 0054, 0055, 0068.

Independent Claim 16 recites a system comprising a detecting unit (19), arranged at a detection point on a transmission channel (15) between a first (13a) and a second (13b) VoIP endpoints. *See, e.g.*, ¶¶ 0064-0068, fig. 3. The detection unit is used to detect an enumeration (19a) of RTP speech packets transmitted from the first (13a) to the second (13b) VoIP endpoints as a first number (19a output line), and to detect an enumeration (19b) of RTP speech packets transmitted from the second (13b) to the first (13a) VoIP endpoints as a second number (19b output line). *See, e.g.*, ¶¶ 0064-0068, fig. 3. The system additionally includes an arithmetic processing unit (23) connected on the input side to the detecting unit (19) where the arithmetic processing unit (23) calculates, from the first (19a output line) and second (19b output line) numbers, a value representing the transmission quality (23 output line). *See, e.g.*, ¶¶ 0050-0052, 0065, fig. 3.

VI. GROUNDS OF REJECTION TO BE REVIEWED ON APPEAL

Claims 1, 2 and 5-23 were rejected by the Examiner under 35 U.S.C. § 112, second paragraph, as being indefinite for failing to particularly point out and distinctly claim the subject matter which applicant regards as the invention. In the Advisory Action mailed February 2, 2009, the Examiner indicated that Appellant's amendments to Claims 1 and 14-16 have overcome this rejection. As noted above, however, Appellant subsequently submitted an amendment to correct an oversight with respect to the § 112 rejection.

Appellant expects that the Examiner will enter the amendment and maintain the conclusion that the § 112 rejection has been overcome.

Claims 1-2, 5, 8-16 and 18-21 were rejected under 35 U.S.C. § 103(a) as being unpatentable over U.S. Patent No. 6,521,746 issued to Israel Elchonin Sand ("*Sand*") in view of U.S. Patent No. 6,678,250 issued to David a Grabelsky et al. ("*Grabelsky*").

Claim 17 was rejected under 35 U.S.C. § 103(a) as being unpatentable over *Sand* and *Grabelsky* and further in view of U.S. Patent No. 6,553,515 issued to Charles J. Gross et al. ("*Gross*").

VII. ARGUMENT

A. Rejection of Claims 1-2, 5, 8-16 and 18-21 under 35 U.S.C. § 103(a) using references *Sand* and *Grabelsky*.

In the Final Office Action mailed December 8, 2008, and in the Advisory Action mailed February 2, 2009, the Examiner states that the combination of *Sand* and *Grabelsky* renders all limitations of all independent claims (Claims 1 and 14-16) obvious. Appellant respectfully disagrees.

The method claims in independent Claims 1 and 14-15 recite:

> detecting . . . an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as <u>a first number</u>, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as <u>a second number</u>;
>
> <u>arithmetically processing the first and second numbers</u>, and outputting <u>a value which is based on the arithmetical processing</u> representing the transmission quality.

The system claim in Independent Claim 16 similarly recites:

> a detecting unit . . . to detect an enumeration of RTP speech packets transmitted from the first to the second VoIP endpoints as <u>a first number</u>, and to detect an enumeration of RTP speech packets transmitted from the second to the first VoIP endpoints as <u>a second number</u>;

> an arithmetic processing unit connected on the input side to the detecting unit to calculate a value representing the transmission quality from the first and second numbers.

These limitations make clear that the determination of transmission quality between two VoIP endpoints is based on a bidirectional RTP speech packet count. The bidirectional nature of this packet count results from the fact that a first number represents an enumeration of RTP speech packets travelling in one direction (i.e., from the first to the second VoIP endpoint), and a second number represents an enumeration of RTP speech packets travelling in the opposite direction (i.e., from the second to the first VoIP endpoint). Merely detecting the first and second numbers, however, does not complete the determination of the transmission quality between the two VoIP endpoints. Rather, the claims are explicit that the transmission quality (i.e., the value that is calculated/output by the arithmetic processing unit or as a result of the arithmetical processing) results from arithmetically processing the first and second numbers. In other words, the first and second numbers—by themselves—do not provide any indication of transmission quality. That indication is only provided by arithmetically processing the two numbers in accordance with Appellant's disclosure. ***The claims are explicit that the final output value, which represents the transmission quality, is calculated based on both the first and second numbers.***

Despite these clear and explicit limitations in Claims 1 and 14-16, the Examiner has concluded that "the features upon which applicant relies (i.e., comparing a first enumeration of packet received by a first gateway from a second gateway with a second enumeration of packets received by the second gateway from the first gateway) are not recited in the rejected claim(s)." (Advisory Action, Page 2 (Continuation Sheet).) Appellant respectfully submits that, contrary to the Examiner's conclusion, these features are explicitly recited in Claims 1 and 14-15 as "arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality," and in Claim 16 as "an arithmetic processing unit connected on the input side to the detecting unit to calculate a value representing the transmission quality from the first and second numbers."

Further, the Examiner has claimed that *Grabelsky* discloses the features described above. However, *Grabelsky* teaches determining the transmission quality of network

communication based only on unidirectional packet counts, while—as just discussed—the present invention determines transmission quality based on bidirectional packet counts. Specifically, *Grabelsky* teaches to compare the received packet count by a first gateway from a second gateway with the expected (or sent) number of packets from the second to the first gateway. (*Grabelsky*, column 7, lines 45-56.) In other words, the second gateway reports the number of packets it received back to the first gateway; and the first gateway, which knows how may packets it sent, can compare the "sent count" with the "received count" to determine a measurement of transmission quality—i.e., the fractional packet loss from the first to the second gateway. (*Grabelsky*, column 7, lines 56-57.) This is further explained as how well the second gateway "hears" the first gateway. (*Grabelsky*, column 7, lines 49-50.) Appellant notes that the resulting measurement of transmission quality is determined solely using unidirectional packet counts (i.e., packets travelling from the first to the second gateway.)

Similarly, and as the Examiner explained in the Advisory Action, *Grabelsky* discloses that this same determination can be made for communications travelling in the opposite direction. (*Grabelsky*, column 7, lines 56-64.) Appellant respectfully submits, however, that this measurement is similarly based solely on a unidirectional packet count, the only difference being that it is made based on packets travelling from the second gateway to the first gateway. (*Id.*) Thus, *Grabelsky* teaches two different transmission quality measurements: from the perspective of the first gateway, these measurements represent (1) a transmit packet quality measurement and (2) a receive packet quality measurement. (*Grabelsky*, column. 7, lines 45-64.) Again, each of these two, separate measurements are based on unidirectional packet counts. Neither rely on bidirectional packet counts.

Further, *Grabelsky* does not teach or suggest the combination of these two, separate transmission quality measurements in order to obtain a quality measurement that is calculated based on both (1) a first enumeration of packets received by a first gateway from a second gateway, and (2) a second enumeration of packets received by the second gateway from the first gateway (together, a bidirectional packet count). Rather, *Grabelsky* simply teaches that the two, separate measurements can be determined. That *Grabelsky* fails to suggest the combination of these two measurements is further evidenced by the fact that *Grabelsky* does

not disclose or teach the equilibrium condition of the bidirectional enumeration of RTF speech packets transmitted between two VoIP endpoints. As Appellant's specification makes clear, Appellant's recognition of this equilibrium condition allows for the disclosed system and methods, which do not require the use of complicated and/or computationally intensive algorithms. Thus, *Grabelsky*'s silence with respect to the equilibrium condition further shows that the measurements therein are based purely on unidirectional packet counts.

Like *Grabelsky*, *Sand* also fails to disclose the limitations discussed above. Since *Sand* and *Grabelsky* fail to teach or suggest detecting, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number; and then arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing and represents the transmission quality, it is respectfully requested that the rejection of independent Claims 1 and 14-16 under 35 U.S.C. § 103(a) be withdrawn. Appellant respectfully submits that the dependent claims are allowable at least to the extent of the independent Claim to which they refer, respectively. Thus, Appellant respectfully requests reconsideration and allowance of the dependent claims.

B. Rejection of Claim17 under 35 U.S.C. § 103(a) using references *Sand*, *Grabelsky*, and *Gross*.

The Examiner has rejected Claim 17 under 35 U.S.C. § 103(a) based on *Sand*, *Grabelsky*, and *Gross*. Claim 17 depends from independent Claim 16. As discussed above, *Sand* and *Grabelsky* fail to disclose all of the limitations of independent Claim 16. *Gross* similarly fails to disclose these limitations. Thus, Appellant respectfully requests reconsideration and allowance of dependent Claim 17.

SUMMARY

Appellant authorizes the Commissioner to charge $540.00 for the Appeal Brief to Deposit Account No. 50-4871 of King & Spalding L.L.P. Appellant believes there are no additional fees due at this time, however, the Commissioner is hereby authorized to charge any fees necessary or credit any overpayment to Deposit Account No. 50-4871 of King & Spalding L.L.P.

Respectfully submitted,
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APPENDIX A - CLAIMS INVOLVED IN APPEAL

1. (Previously Presented) A method for testing the transmission quality of a bidirectional real speech transmission or multicast connection over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting RTP speech packets from the first to the second VoIP endpoints, and transmitting RTP speech packets from the second to the first VoIP endpoints;

detecting, at a detection point on a transmission channel between the first and the second VoIP endpoints, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number; and

arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality.

2. (Previously Presented) The method as claimed in claim 1, wherein the predetermined time period for a 10 Mbit/s transmission channel is longer that 5 s.

3-4. (Cancelled)

5. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is subjected to a threshold value discrimination to suppress side effects due to features of a communication protocol.

6. (Previously Presented) The method as claimed in claim 22, wherein quotients outside a predetermined tolerance range around the value 1 are valid as a representation of a reduced transmission quality.

7. (Previously Presented) The method as claimed in claim 23, wherein difference values outside a predetermined tolerance range around the value 0 are valid as a representation of a reduced transmission quality.

8. (Previously Presented) The method as claimed in claim I, wherein the detected first and second numbers and/or the ealculated values for a plurality of first and second VoIP endpoints connected to the IP network between which bidirectional speech connections exist in each case are logged.

9. (Previously Presented) The method as claimed in claim 1, wherein the detected first and second numbers and/or the calculated values for a plurality of first and second VoIP endpoints connected to the IP network within which bidirectional speech connections exist in each case are subjected to summarizing statistical processing to obtain an overall value representing the overall transmission quality of the IP network or of a section of the overall transmission quality of the IP Network.

10. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is signaled to subscribers at the first and/or second VoIP endpoints and/or to an operation control center of the IP network.

11. (Previously Presented) The method as claimed in claim I, wherein the value representing the transmission quality is used as an input variable for controlling the speech transmission over the IP network.

12. (Previously Presented) The method as claimed in claim 1, wherein the value representing the transmission quality is determined substantially in real time and is signaled or is used as an input variable for controlling the speech transmission.

13. (Previously Presented) The method according to claim 2, wherein the predetermined time period is in the range of about 10 s to 30 s.

14. (Previously Presented) A method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting RTP speech packets from the first to the second VoIP endpoints, and transmitting RTP speech packets from the second to the first VoIP endpoints;

detecting, at a detection point on a transmission channel between the first and the second VoIP endpoints, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number;

arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality; and

routing a connection between the first and second VoIP endpoints based on the value.

15. (Previously Presented) A method for controlling a speech transmission over an IP network between a first VoIP endpoint and a second VoIP endpoint, comprising:

transmitting RTP speech packets from the first to the second VoIP endpoints,

transmitting RTP speech packets from the second to the first VoIP endpoint;

detecting, at a detection point on a transmission channel between the first and the second VoIP endpoints, over a predetermined time period, an enumeration of the transmitted RTP speech packets from the first to the second VoIP endpoints as a first number, and an enumeration of the transmitted RTP speech packets from the second to the first VoIP endpoints as a second number;

arithmetically processing the first and second numbers, and outputting a value which is based on the arithmetical processing representing the transmission quality; and

setting transmission parameters based on the value.

16. (Previously Presented) A system, comprising:

a detecting unit, arranged at a detection point on a transmission channel between a first and a second VoIP endpoints, to detect an enumeration of RTP speech packets transmitted from the first to the second VoIP endpoints as a first number, and to detect an enumeration of RTP speech packets transmitted from the seeond to the first VoIP endpoints as a second number;

an arithmetic processing unit connected on the input side to the detecting unit to calculate a value representing the transmission quality from the first and second numbers.

17. (Previously Presented) The system as claimed in claim 16, wherein the arithmetic processing unit has a division or subtraction stage.

18. (Previously Presented) The system as claimed in claim 16, wherein connected downstream of the arithmetic processing unit is a threshold value discriminator to evaluate the value representing the transmission quality with the aid of at least one predetermined threshold value.

19. (Previously Presented) The system as claimed in claim 16, further comprising a storage device connected on the input side to the output of the detecting device and/or of the arithmetic processing unit to log the first and second numbers and/or the calculated values.

20. (Previously Presented) The system as claimed in claim 16, further comprising a statistical processing unit, connected on the input side to the output of the detecting device and/or of the arithmetic processing unit, to summarize statistical processing of the detected numbers or calculated values in order to evaluate the overall transmission quality of the IP network or of a section of the same.

21. (Previously Presented) The system as claimed in claim 16, further comprising a signaling device to signal the calculated value or the overall value to the subscribers at the first and/or second VoIP endpoint and/or to an operation control center of the IP network.

22. (Previously Presented) The method as claimed in claim 1, wherein the arithmetic processing includes a division, where a value 1 of the quotient represents the highest transmission quality.

23. (Previously Presented) The method as claimed in claim 1, wherein the arithmetic processing includes a subtraction, where a value 0 for the difference represents the highest transmission quality.

APPENDIX B - EVIDENCE

NONE

APPENDIX C: RELATED PROCEEDINGS

NONE